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EXAMINER	
YUEN, KAN	

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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

a

Office Action Summary	Application No. 10/777,933	Applicant(s) BEIGHTOL ET AL.
	Examiner Kan Yuen	Art Unit 2616

-- *The MAILING DATE of this communication appears on the cover sheet with the correspondence address* --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on 30 January 2007.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) 1-44 is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 1-44 is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on 12 February 2004 is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application

6) ☐ Other: _____.

Detailed Action

Claim Objections

1. Claim 29 is objected to because of the following informalities:

The applicant is suggested to spell out the term "TTY/TDD". Appropriate correction is required.

Claim Rejections - 35 USC § 102

2. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (e) the invention was described in (1) an application for patent, published under section 122(b), by another filed in the United States before the invention by the applicant for patent or (2) a patent granted on an application for patent by another filed in the United States before the invention by the applicant for patent, except that an international application filed under the treaty defined in section 351(a) shall have the effects for purposes of this subsection of an application filed in the United States only if the international application designated the United States and was published under Article 21(2) of such treaty in the English language.

3. Claim 30 is rejected under 35 U.S.C. 102(e) as being anticipated by McNiff et al. (Pat No.: 6807150).

In claim 30, Mcniff et al. disclosed the method of transmitting data packets from a first communication device to a second communication device via the IP-based network using a first transmission protocol that does not retransmit transmitted packets that are at least one of lost and damaged (see fig. 1, and fig. 2, column 1, lines 40-67, column 2, lines 1-20). The caller A 18 is the originating device, and Caller B 18 is the destination device. The switch 12 is the port circuit for transmitting information to the destination device. The packet network 14 is the internet. As shown in fig. 2, the call control module

30 monitors the network quality and it also performs switching between transmission protocols based on certain conditions, and these conditions are packet loss measurements, packet delay measurements, and bit error rate measurements. Some transmission protocols are TCP, IP, UDP, etc. As we already know that TCP is used for retransmission, and UDP does not perform retransmission. Therefore we can consider during transmission using UDP that the UDP does not retransmit lost or damaged packets; determining that network performance of the IP-based network is insufficient to transmit quality data signals using the first transmission protocol (see fig. 1, and fig. 2, column 1, lines 40-67, column 2, lines 1-20). The module 30 monitors the quality metric of the communication session and switch the session if necessary; and changing from transmitting data packets using a first transmission protocol to transmitting data packets using a second transmission protocol that provides for retransmission of transmitted packets that are at least one of lost and damaged (see fig. 2, module 30, column 2, lines 40-67, column 3, 1-67, column 5, lines 1-67). As shown in fig. 2, the call control module 30 monitors the network quality and it also performs switching between transmission protocols based on certain conditions, and these conditions are packet loss measurements, packet delay measurements, and bit error rate measurements. Some transmission protocols are TCP, IP, UDP, etc. As we already know that TCP is used for retransmission.

Claim Rejections - 35 USC § 103

4. The factual inquiries set forth in *Graham* v. *John Deere Co.*, 383 U.S. 1, 148 USPQ 459 (1966), that are applied for establishing a background for determining obviousness under 35 U.S.C. 103(a) are summarized as follows:

1. Determining the scope and contents of the prior art.
2. Ascertaining the differences between the prior art and the claims at issue.
3. Resolving the level of ordinary skill in the pertinent art.
4. Considering objective evidence present in the application indicating obviousness or nonobviousness.

5. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

6. Claims 1, 2, 5, 6, 8, 9, 11, 12, 15, 16, 18, 19, 21, 31, 34, 35, 37, 38 are rejected under 35 U.S.C. 103(a) as being unpatentable over McNiff et al. (Pat No.: 6807150), in view of He et al. (Pub No.: 2004/0057456).

For claim 1, Mcniff et al. disclosed the method of a port circuit for transmitting data packets, containing encoded speech signals received from an associated originating device, to the destination device via the IP-based network (see fig. 1, and fig. 2, column 1, lines 40-67, column 2, lines 1-20). The caller A 18 is the originating device, and Caller B 18 is the destination device. The switch 12 is the port circuit for transmitting information to the destination device. The packet network 14 is the internet; transmit buffer mean, connected to the port circuit associated with the originating device, for storing a plurality of the data packets received from the associated

originating device (see fig. 2, memory 42, column 5, lines 1-15). The memory 42 maintains a session information table 44; network activation means for activating the IP-based network to operate using a packet transmission protocol that does not retransmit transmitted packets that are lost or damaged (see fig. 2, module 30, column 2, lines 40-67, column 3, 1-67, column 5, lines 1-67). As shown in fig. 2, the call control module 30 monitors the network quality and it also performs switching between transmission protocols based on certain conditions, and these conditions are packet loss measurements, packet delay measurements, and bit error rate measurements. Some transmission protocols are TCP, IP, UDP, etc. As we already know that TCP is used for retransmission, and UDP does not perform retransmission. Therefore we can consider during transmission using UDP that the UDP does not retransmit lost or damaged packets.

However, Mcniff et al. did not disclose the method of packet retransmission means, operable independently of the packet transmission protocol and responsive to a transmitted packet being lost or damaged, for activating the port circuit to retrieve the packet from the transmit buffer means for retransmission to the destination device. He et al. from the same or similar fields of endeavor teaches the method of packet retransmission means, operable independently of the packet transmission protocol and responsive to a transmitted packet being lost or damaged, for activating the port circuit to retrieve the packet from the transmit buffer means for retransmission to the destination device (see fig. 2, paragraphs 0040, 0041). The RTP layer engine 154 is the packet retransmission means for activating the memory to retransmitting lost packet

based on the RTCP feedback, wherein the feedback activates the retransmission. Thus, it would have been obvious to the person of ordinary skill in the art at the time of the invention to use the method as taught by He et al. in the network of Mcniff et al. The motivation for using the method as taught by He et al. in the network of Mcniff et al. being that it provides reliability in voice transmission network.

Regarding claim 2, He et al. disclosed the method of packet error detection means, connected to the destination device, for generating an indication that identifies a missing packet; and means for transmitting a signal to the port circuit associated with the originating device requesting retransmission of the identified packet (see fig. 2, paragraph 0032, RTCP 210). The RTP layer engine 154 is the packet retransmission means for activating the memory to retransmitting lost packet based on the RTCP feedback, wherein the feedback activates the retransmission.

Regarding claim 5, Mcniff et al. disclosed the method of application detection means for determining that the communication connection serves a speech-based application that requires high quality audio signals (see fig. 1, and fig. 2, column 1, lines 40-67, column 2, lines 1-20).

Regarding claim 6, Mcniff et al. disclosed the method of network control means, responsive to the application detection means, for activating the IP-based transmission medium to transmit the high quality digital encoded speech signals without transcoding (see fig. 1 (call control module 30), and fig. 2, column 1, lines 40-67, column 2, lines 1-20).

Regarding claim 8, Mcniff et al. disclosed the method of destination device identification means for determining presence of a destination device on the communication connection that requires high quality audio signals (see fig. 2, module 30, column 2, lines 40-67, column 3, 1-67, column 5, lines 1-67). The call module 30 is the destination device identification mean.

Regarding claim 9, Mcniff et al. disclosed the method of registration process detection means for determining presence of a subscriber identification process at the destination device (see fig. 2, module 30, column 2, lines 40-67, column 3, 1-67, column 5, lines 1-67).

Claim 11 is rejected similar to claim 1.

Regarding claim 12, He et al. disclosed the method of generating an indication that identifies a missing packet; and transmitting a signal to the port circuit associated with the originating device requesting retransmission of the identified packet (see fig. 2, paragraphs 0040, 0041). The RTP layer engine 154 is the packet retransmission means for activating the memory to retransmitting lost packet based on the RTCP feedback, wherein the feedback activates the retransmission.

Regarding claim 15, Mcniff et al. disclosed the method of determining that the communication connection serves a speech-based application that requires high quality audio signals (see fig. 1, and fig. 2, column 1, lines 40-67, column 2, lines 1-20).

Regarding claim 16, Mcniff et al. disclosed the method of activating, in response to the step of determining, the IP-based transmission medium to transmit the high quality digital encoded speech signals without transcoding (see fig. 1 (call control module 30), and fig. 2, column 1, lines 40-67, column 2, lines 1-20).

Regarding claim 18, Mcniff et al. disclosed the method of determining presence of a destination device on the communication connection that requires high quality audio signals (see fig. 1, and fig. 2, column 1, lines 40-67, column 2, lines 1-20).

Regarding claim 19, Mcniff et al. disclosed the method of determining presence of a subscriber identification process at the destination device (see fig. 2, module 30, column 2, lines 40-67, column 3, 1-67, column 5, lines 1-67). The call module 30 is the destination device identification mean.

Regarding claim 21, Mcniff et al. disclosed the method of managing buffering and subsequent delivery of data to the receiving device, such that pauses in the data

delivery occur at locations that would have a minimal impact on the performance of the receiving device (Mcniff et al. see fig. 2, module 30, column 2, lines 40-67, column 3, 1-67, column 5, lines 1-67). As shown in fig. 2, the call control module 30 monitors the network quality and it also performs switching between transmission protocols based on certain conditions, and these conditions are packet loss measurements, packet delay measurements, and bit error rate measurements. Some transmission protocols are TCP, IP, UDP, etc. As we already know that TCP is used for retransmission, and UDP does not perform retransmission. Therefore we can consider during transmission using UDP that the UDP does not retransmit lost or damaged packets.

However, Mcniff et al. did not disclose the method of detecting when a pause in data delivery is necessary and determining an effect that pauses at specific points in the delivery of data would have on performance of the receiving device. He et al. from the same or similar fields of endeavor teaches the method of detecting when a pause in data delivery is necessary (paragraph 0018). The automatically generated RTCP feedback will be examined to see if any retransmission is necessary; determining an effect that pauses at specific points in the delivery of data would have on performance of the receiving device (paragraph 0018). Data may be retransmitted based on the automatically generated RTCP feedback, which indicates the sequence number of packets. Thus, it would have been obvious to the person of ordinary skill in the art at the time of the invention to use the method as taught by He et al. in the network of Mcniff et al. The motivation for using the method as taught by He et al. in the network of Mcniff et al. being that it provides reliability in voice transmission network.

Regarding claim 31, He et al. disclosed the method of generating an indication that identifies a missing packet; and transmitting a signal to the first communication device requresting retransmission of the identified packet (see fig. 2, paragraph 0032, RTCP 210). The RTP layer engine 154 is the packet retransmission means for activating the memory to retransmitting lost packet based on the RTCP feedback, wherein the feedback activates the retransmission.

Regarding claim 34, Mcniff et al. disclosed the method of determining that the communication connection serves a speech-based application that requires high quality audio signals (see fig. 1, and fig. 2, column 1, lines 40-67, column 2, lines 1-20).

Regarding claim 35, He et al. disclosed the method of activating, in response to the step of determining, the IP-based network to transmit the high quality digital encoded speech signals without transcoding (see fig. 1 (call control module 30), and fig. 2, column 1, lines 40-67, column 2, lines 1-20).

Regarding claim 37, Mcniff et al. disclosed the method of determining presence of a second communication device on the communication connection that requires high quality audio signals (see fig. 2, module 30, column 2, lines 40-67, column 3, 1-67, column 5, lines 1-67). The call module 30 is the destination device identification mean.

Regarding claim 38, He et al. disclosed the method of determining presence of a subscriber identification process at the second communication device (see fig. 2, module 30, column 2, lines 40-67, column 3, 1-67, column 5, lines 1-67).

7. Claims 40-42, 44 are rejected under 35 U.S.C. 103(a) as being unpatentable over McNiff et al. (Pat No.: 6807150), in view of Lanzafame et al. (Pub No.: 2003/0026275).

For claim 40, Mcniff et al. disclosed the method of providing a transmit buffer for temporary storage of data packets to be sent across an IP-based communication network (fig. 2, memory 42, column 5, lines 1-67). Each switch 12 comprises a memory 42 for storing information to be sent across the internet network; storing in the transmit buffer at least one data packet that is transmitted across the IP-based communication network (fig. 2, memory 42, column 5, lines 1-67). Each switch 12 comprises a memory 42 for storing information to be sent across the internet network. However, Mcniff et al. did not disclose the method of varying a size of the transmit buffer based on input from at least one of the IP-based communication network and a destination system which is on the communication connection and connected to the IP-based communication network. Lanzafame et al. from the same or similar fields of endeavor teaches the method of varying a size of the transmit buffer based on input from at least one of the IP-based communication network and a destination system which is on the communication connection and connected to the IP-based communication network (paragraph 0031). The buffer size is adjusted based on the analysis of the received voice signal. Thus, it would have been obvious to the person of ordinary skill in the art at the time of the invention to use the method as taught by Lanzafame et al. in the network of Mcniff et al. The motivation for using the method as taught by Lanzafame et al. in the

network of Mcniff et al. being that it provides able to adjust memory sizes with control signal.

Regarding claim 41, Mcniff et al. disclosed the method of transmission delay in the IP-based communication network, speed of processing received data packets, time required to identify absence of a data packet in a sequence of received data packets, and time required to receive a transmitted data packet (fig. 2, memory 42, column 5, lines 1-67).

Regarding claim 42, Mcniff et al. disclosed the method of transmitting data packets from the transmit buffer to the destination system via the IP-based network using a first transmission protocol that does not retransmit transmitted packet that are at least one of lost and damaged (see fig. 1, and fig. 2, column 1, lines 40-67, column 2, lines 1-20). The caller A 18 is the originating device, and Caller B 18 is the destination device. The switch 12 is the port circuit for transmitting information to the destination device. The packet network 14 is the internet. As shown in fig. 2, the call control module 30 monitors the network quality and it also performs switching between transmission protocols based on certain conditions, and these conditions are packet loss measurements, packet delay measurements, and bit error rate measurements. Some transmission protocols are TCP, IP, UDP, etc. As we already know that TCP is used for retransmission, and UDP does not perform retransmission. Therefore we can consider during transmission using UDP that the UDP does not retransmit lost or damaged packets; determining that network performance of the IP-based network is insufficient to transmit quality data signals using the first transmission protocol (see fig. 1, and fig. 2,

column 1, lines 40-67, column 2, lines 1-20). The module 30 monitors the quality metric of the communication session and switch the session if necessary; and changing from transmitting data packets using the first transmission protocol to transmitting data packets using a second transmission protocol that provides for retransmission of transmitted packets that are at least one of lost and damaged (see fig. 2, module 30, column 2, lines 40-67, column 3, 1-67, column 5, lines 1-67). As shown in fig. 2, the call control module 30 monitors the network quality and it also performs switching between transmission protocols based on certain conditions, and these conditions are packet loss measurements, packet delay measurements, and bit error rate measurements. Some transmission protocols are TCP, IP, UDP, etc. As we already know that TCP is used for retransmission.

Regarding claim 44, Mcniff et al. disclosed the method of determining presence of a destination system on the communication connection that requires high quality audio signals (see fig. 1, and fig. 2, column 1, lines 40-67, column 2, lines 1-20).

8. Claims 3, 4, 13, 14, 32, 33, 43 are rejected under 35 U.S.C. 103(a) as being unpatentable over McNiff et al. (Pat No.: 6807150), in view of He et al. (Pub No.: 2004/0057456), as applied to claim 1 above, and further in view of Lanzafame et al. (Pub No.: 2003/0026275).

For claim 3, Mcniff et al. and He et al. did not disclose the method of transmit buffer control means for transmitting a signal to the port circuit associated with the

originating device to regulate size of the transmit buffer. Lanzafame et al. from the same or similar fields of endeavor teaches the method of transmit buffer control means for transmitting a signal to the port circuit associated with the originating device to regulate size of the transmit buffer (paragraph 0031). The buffer size is adjusted based on the analysis of the received voice signal. Thus, it would have been obvious to the person of ordinary skill in the art at the time of the invention to use the method as taught by Lanzafame et al. in the network of Mcniff et al. The motivation for using the method as taught by Lanzafame et al. in the network of Mcniff et al. being that it provides able to adjust memory sizes with control signal.

Regarding claim 4, Lanzafame et al. disclose the method of jitter buffer management means for regulating a size of a jitter buffer associated with the destination device as a function of at least one of network transmission delay, speed of processing received packets, time required to identify absence of a packet in a sequence of received packet, and time required to receive a retransmitted packet (paragraph 0031). The buffer size is adjusted based on the analysis of the received voice signal.

Regarding claim 13, Lanzafame et al. disclosed the method of transmitting a signal to the port circuit associated with the originating device to regulate size of the transmit buffer (paragraph 0031). The buffer size is adjusted based on the analysis of the received voice signal.

Regarding claim 14, Lanzafame et al. disclosed the method regulating a size of a jitter buffer associated with the destination device as a function of at least one of: network transmission delay, speed of processing received packets, time required to

identify absence of a packet in a sequence of received packets, and time required to receive a retransmitted packet (paragraph 0031). The buffer size is adjusted based on the analysis of the received voice signal

Regarding claim 32, Lanzafame et al. disclosed the method transmitting a signal to the first communication device to regulate size of the transmit buffer (paragraph 0031). The buffer size is adjusted based on the analysis of the received voice signal.

Regarding claim 33, Lanzafame et al. disclosed the method regulating a size of a jitter buffer associated with the second communication device as a function of at least one of: network transmission delay, speed of processing received packets, time required to identify absence of a packet in a sequence of received packets, and time required to receive a retransmitted packet (paragraph 0031). The buffer size is adjusted based on the analysis of the received voice signal.

Regarding claim 43, He et al. disclosed the method of generating, at the destination system, an indication that identifies a missing packet; and transmitting a signal to the transmit buffer requesting retransmission of the identified packet (see fig. 2, paragraphs 0040, 0041). The RTP layer engine 154 is the packet retransmission means for activating the memory to retransmitting lost packet based on the RTCP feedback, wherein the feedback activates the retransmission.

9. Claims 7, 10, 17, 20, 36, 39 are rejected under 35 U.S.C. 103(a) as being unpatentable over McNiff et al. (Pat No.: 6807150), in view of He et al. (Pub No.:

2004/0057456), as applied to claim 1 above, and further in view of Yazdy et al. (Pub No.: 2004/0017806).

For claim 7, Mcniff et al. and He et al. both did not disclose the method of process disabling means, responsive to conclusion of operation of the speech-based application, for disabling operation of the packet retransmission means. Yazdy et al. from the same or similar fields of endeavor teaches the method of process disabling means, responsive to conclusion of operation of the speech-based application, for disabling operation of the packet retransmission means (paragraph 0013). Thus, it would have been obvious to the person of ordinary skill in the art at the time of the invention to use the method as taught by Yazdy et al. in the network of Mcniff et al. The motivation for using the method as taught by Yazdy et al. in the network of Mcniff et al. being that it provides able to increase quality transmission using higher layer protocol.

Regarding claim 10, Yazdy et al. disclosed the method of process disabling means, responsive to conclusion of operation of the subscriber identification process, for disabling operation of the packet retransmission means (paragraph 0013).

Regarding claim 17, Yazdy et al. disclosed the method of disabling, in response to conclusion of operation of the speech-based application, operation of the step of activating the port circuit to retransmit transmitted packets that are lost or damaged (paragraph 0013).

Regarding claim 20, Yazdy et al. disclosed the method of disabling, in response to conclusion of operation of the subscriber identification process, operation of the step

of activating the port circuit to retransmit transmitted packets that are lost or damaged (paragraph 0013).

Regarding claim 36, Yazdy et al. disclosed the method of disabling, in response to the conclusion of opera6on of the speech-based application, operation of the step of activating the first communication device to retransmit transmitted packers that are lost or damaged (paragraph 0013).

Regarding claim 39, Yazdy et al. disclosed the method of disabling, in response to conclusion of operation of the subscriber identification process, operation of the step of activating the port circuit to retransmit transmitted packets that are lost or damaged (paragraph 0013).

10. Claims 22-28 are rejected under 35 U.S.C. 103(a) as being unpatentable over McNiff et al. (Pat No.: 6807150), in view of He et al. (Pub No.: 2004/0057456), as applied to claim 1 above, and further in view of Mesiwala (Pub No.: 2002/0027880).

For claim 22, Mcniff et al and He et al. both did not disclose the method of the data consist of speech inputs to an automatic speech recognition resource; and the step of managing the buffering implements pauses in the delivery of speech to the resource between words rather than within words. Mesiwala from the same or similar fields of endeavor teaches the method of the data consist of speech inputs to an automatic speech recognition resource; and the step of managing the buffering implements pauses in the delivery of speech to the resource between words rather than within words (paragraph 0045). Thus, it would have been obvious to the person of ordinary

skill in the art at the time of the invention to use the method as taught by Mesiwala in the network of Mcniff et al. and He et al. The motivation for using the method as taught by Mesiwala in the network of Mcniff et al. and He et al. being that it provides able to increase quality transmission using higher layer protocol.

Regarding claim 23, Mesiwala disclosed the method of data consist of speech inputs to an automatic speech recognition resource; and the step of managing the buffering implements pauses in the deliver/of speech to the resource between phrases rather than within phrases (paragraph 0045).

Regarding claim 24, Mesiwala disclosed the method of the data consist of speech inputs to an automatic speech recognition resource; and the step of managing the buffering implements pauses in the delivery of speech to the resource between commands rather than within commands (paragraph 0045).

Regarding claim 25, Mesiwala disclosed the method of the data consist of speech inputs to a voice-recording resource; and the step of managing the buffering implements pauses in the delivery of speech to the resource between words rather than within words (paragraph 0045).

Regarding claim 26, Mesiwala disclosed the method of the data consist of speech inputs to a voice-recording resource; and the step of managing the buffering implements pauses in the delivery of speech to the resource between phrases rather than within phrases (paragraph 0045).

Regarding claim 27, Mesiwala disclosed the method of the data consist of tonal inputs to a tone-dejection resource; and the step of managing the buffering implements

pauses in the delivery of audio tones to the resource between tones, rather than within tones (paragraph 0045).

Regarding claim 28, Mesiwala disclosed the method of the data consist of audio signals in which the duration of individual signals is important; and the step of managing the buffering implements pauses in the delivery of signals to the resource which do not occur within time-sensitive signal components (paragraph 0045).

11. Claim 29 is rejected under 35 U.S.C. 103(a) as being unpatentable over McNiff et al. (Pat No.: 6807150), in view of He et al. (Pub No.: 2004/0057456), as applied to claim 1 above, and further in view of Bostrom et al. (Pub No.: 2005/0047402).

For claim 29, Mcniff et al. and he et al. both did not disclose the method of the data consist of TTY/TDD characters; and the step of managing the buffering implements pauses in the delivery of characters to the resource between individual characters rather than within characters. Bostrom et al. from the same or similar fields of endeavor teaches the method of the data consist of TTY/TDD characters; and the step of managing the buffering implements pauses in the delivery of characters to the resource between individual characters rather than within characters (paragraph 0016). Thus, it would have been obvious to the person of ordinary skill in the art at the time of the invention to use the method as taught by Bostrom et al. in the network of Mcniff et al. and He et al. The motivation for using the method as taught by Bostrom et al. in the

network of Mcniff et al. and He et al. being that it provides able to increase quality transmission using higher layer protocol.

Conclusion

12. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure. Cheng et al. (Pub No.: 2003/0133408), is show systems which considered pertinent to the claimed invention.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Kan Yuen whose telephone number is 571-270-1413. The examiner can normally be reached on Monday-Friday 10:00a.m-3:00p.m EST.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Ricky O. Ngo can be reached on 571-272-3139. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

KY

RICKY Q. NGO
SUPERVISORY PATENT EXAMINER